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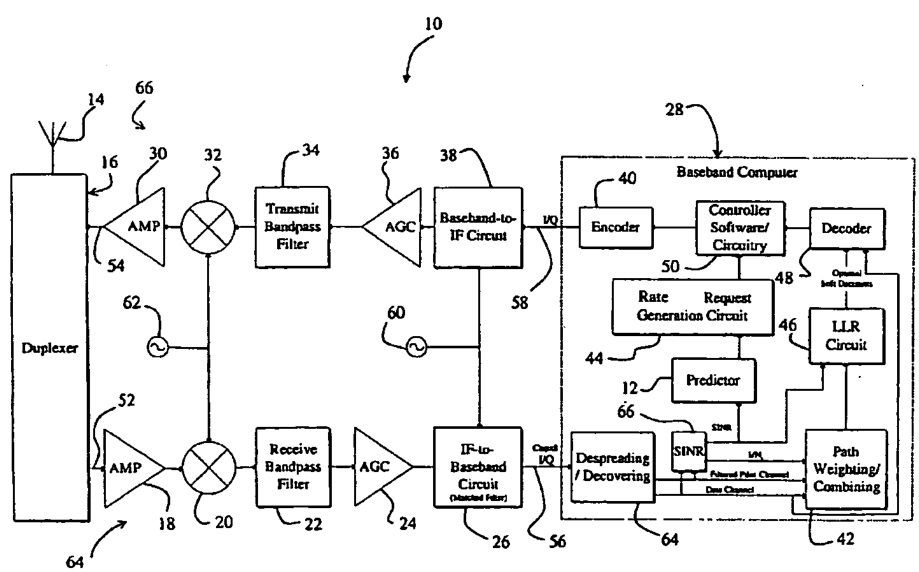

(57) Abstract: A system (10) for providing an accurate prediction of a signal-to-interference noise ratio. The system (10) includes a first circuit for receiving a signal transmitted across a channel via an external transmitter. A second circuit generates a sequence of estimates of signal-to-interference noise ratio based on the received signal. A third circuit determines a relationship between elements of the sequence of estimates. A fourth circuit employs the relationship to provide a signal-to-interference noise ratio prediction for a subsequently received signal.

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SYSTEM AND METHOD FOR ACCURATELY PREDICTING SIGNAL TO INTERFERENCE AND NOISE RATIO TO IMPROVE COMMUNICATIONS SYSTEM PERFORMANCE

BACKGROUND OF THE INVENTION

I. Field of Invention:

This invention relates to communications systems. Specifically, the present invention relates to systems for predicting the signal to interference and noise ratio (SINR) of a received signal to facilitate data rate control in wireless communications systems.

II. Description of the Related Art:

Wireless communications systems are used in a variety of demanding applications including search and rescue and business applications. In addition, wireless communications systems are increasingly employed to transfer computer data in office network and Internet applications. Such applications require efficient and reliable communications systems that can effectively operate in electrically fading and noisy environments and that can handle high data transfer rates.

Cellular telecommunications systems are characterized by a plurality of mobile stations (e.g. cellular telephones or wireless phones) in communication with one or more base stations. The communications link from a base station to a mobile station is the forward link. The communications link from the mobile station to the base station is the reverse link.

Signals transmitted by a mobile station are received by a base station and often relayed to a mobile switching center (MSC). The MSC in turn routes the signal to a public switched telephone network (PSTN) or to another mobile station. Similarly, signals are often transmitted from the public switched telephone network to a mobile station via a base station and a

mobile switching center. Each base station governs a cell, a region within which a mobile station may communicate via the base station.

In typical mobile communication systems, information is encoded, modulated, and transmitted over a channel and received, demodulated and decoded by a receiver. In many modern communication systems, such as Code Division Multiple Access (CDMA) cellular networks, the information is encoded digitally for channel noise, capacity, and data security reasons. A convolutional encoder or turbo encoder often performs the encoding of the information.

As is well known in the art, a convolutional encoder converts a sequence of input data bits to a codeword based on a convolution of the input sequence with itself or with another signal. Code rate and generating polynomials are used to define a convolutional code. Convolutional encoding of data combined with a Viterbi decoder is a well-known technique for providing error correction coding and decoding of data. Turbo encoders employ turbo codes, which are serial or parallel concatenations of two or more constituent codes such as convolutional codes.

Mobile communications systems are typified by the movement of a receiver relative to a transmitter or vice versa. The communications link between transmitters and receivers in a mobile communications system is a fading channel. Mobile satellite communications systems, having a transmitter on a spacecraft and a receiver on a ground based vehicle, cellular telephone systems and terrestrial microwave systems are examples of fading communications systems. A fading channel is a channel that is severely degraded. The degradation results from numerous effects including multipath fading, severe attenuation due to the receipt via multiple paths of reflections of the transmitted signal off objects and structures in the atmosphere and on the surface, and from interference caused by other users of the communications system. Other effects contributing to the impairment of the faded channel include Doppler shift due to the movement of the receiver relative to the transmitter and additive noise.

Typically, an information signal is first converted into a form suitable for efficient transmission over the channel. Conversion or modulation of the information signal involves varying a parameter of a carrier wave on the basis of the information signal in such a way that the spectrum of the resulting modulated carrier is confined within the channel bandwidth. At a user location, the original message signal is replicated from a version of the modulated carrier received subsequent to propagation over the channel. Such replication is generally achieved by using an inverse of the modulation process employed by the source transmitter.

In a CDMA system, all frequency resources are allocated simultaneously to all users of the cellular network. Each user employs a noise-like wide band signal occupying the entire frequency allocation. The encoder facilitates the encoding of necessary redundant data within each transmission frame to take advantage of the entire frequency allocation, and also facilitates the variable rate transmission on a frame by frame basis.

For voice communication, the capacity of a CDMA system is maximized by having each user transmit only as much data as is necessary. This is because each user's transmission contributes incrementally to the interference in a CDMA communication system. A very effective means of reducing each user's burden on capacity without reducing the quality of service to that user is by means of variable rate transmission. The use of a variable rate communication channel reduces mutual interference by eliminating unnecessary transmissions when there is no useful speech to be transmitted.

Due to the characteristics of voice communication, power control is typically utilized in a CDMA system to guarantee each user a reliable link for certain fixed data rates. Vocoder can provide variable rate source coding of speech data, using the technique described in U.S. Patent No. 5,414,796, May 9, 1995, entitled "Variable Rate Vocoder". Once vocoder generates a sequence of information bits at certain rate, power control will try to adjust the user to transmit as less power as possible that can reliably support the rate. Power control, thus by suppressing each user's contribution to the total interference,

facilitates the maximum capacity of a CDMA voice system in sense that the number of active users is maximized.

For data communication, the parameters, which measure the quality and effectiveness of a system, are the transmission delay required for transferring a data packet and the average throughput rate of the system. Transmission delay is an important metric for measuring the quality of the data communication system. The average throughput rate is a measurement of the efficiency of the data transmission capacity of the communication system. In order to optimize the above parameters for a data communication system, rate control, instead of power control, is typically utilized. The above differences between the voice and data communication systems can be better understood by the following different characteristics between the voice and data communications.

A significant difference between voice services and data services is the fact that the former imposes stringent and fixed delay requirements. Typically, the overall one-way delay of speech frames must be less than 100 msec. In contrast, the data delay can become a variable parameter used to optimize the efficiency of the data communication system. Specifically, more efficient error correcting coding techniques that require significantly larger delays than those that can be tolerated by voice services can be utilized. An exemplary efficient coding scheme for data is disclosed in U.S. Patent Application Serial No. 08/743,688, entitled "SOFT DECISION OUTPUT DECODER FOR DECODING CONVOLUTIONALLY ENCODED CODEWORDS", filed November 6, 1996, assigned to the assignee of the present invention and incorporated by reference herein.

Another significant difference between voice services and data services is that the former requires a fixed and common grade of service (GOS) for all users. Typically, for digital systems providing voice services, this translates into a fixed and equal transmission rate for all users and a maximum tolerable value for the error rates of the speech frames. In contrast, for data services, the GOS can be different from user to user and can be a parameter optimized to increase the overall efficiency of the data communication system. The GOS

of a data communication system is typically defined as the total delay incurred in the transfer of a predetermined amount of data, hereinafter referred to as a data packet.

Yet another significant difference between voice services and data services is that the former requires a reliable communication link which, in the exemplary CDMA communication system, is provided by soft handoff. Soft handoff results in redundant transmissions from two or more base stations to improve reliability. However, this additional reliability is not required for data transmission because the data packets received in error can be retransmitted. For data services, the transmit power used to support soft handoff can be more efficiently used for transmitting additional data. A method and apparatus which is optimized for the wireless transmission of digital data is described in U.S. Patent Application Serial No. 08/963,386 entitled "Method and Apparatus For Higher Rate Packet Data Transmission", which is assigned to the assignee of the present invention and incorporated by reference herein.

As a conclusion of the above characteristics of data communication, a data communication system designed to optimize the average throughput will attempt to serve each user from the best serving base station and at the highest data rate R_b which the user can reliably support. The above conclusion is disclosed in U.S. Patent Application Serial No. 08/963,386 entitled "Method and Apparatus For Higher Rate Packet Data Transmission", which is assigned to the assignee of the present invention and incorporated by reference herein. As a result of the above conclusion, in the modern high-data-rate (HDR) system, base station transmits always at maximum power to only one user at each time slot and uses rate control to adjust the maximum rate that the user can reliably receive. As a characteristic of data communication, throughput is more important to the forward link than reverse link.

A proper rate control algorithm contains 2 loops, inner loop and outer loop. The inner loop controls the forward-link data rate based on the difference between the average SINR of the next packet and the SINR thresholds of all the data rates, while the outer loop adjusts the SINR

thresholds of the data rats based on the forward link PER. For convenience, the average SINR of a packet and the SINR thresholds of all data rats will be refereed to as *packet SINR* and *SINR thresholds*, respectively.

The SINR thresholds reflect the performance of the modem design, but are mainly determined by the channel statistics. We expect that the SINR thresholds change slowly with relatively small variances, thus a tracking loop based on PER will achieve good performance. Further details and analysis on how the outer loop can be done is out of the scope of this study.

In this patent, we assume that the SINR thresholds are fixed. We will focus on the design of the inner loop algorithm. The core technique inside the inner loop is channel prediction.

In HDR system, forward-link traffic channels support 11 data rates, each data rate corresponding to a deterministic packet length associated with 1, 2, 4, 8 or 16 slots. Some packet lengths can support multiple rates. Typically, higher rates are associated with shorter packet lengths.

The predictor will predict the next packet SINR for all packet lengths. The mobile will attempt to request the highest rate by comparing the predictions with the SINR thresholds. For convenience, the prediction of the next packet SINR for a given packet length will be simply refereed to as *prediction*.

In the HDR system, the data rate request information is sent to the BS over the reverse-link data rate control (DRC) channel once every slot. The BS includes a scheduler that schedules forward link traffic packets in accordance with a fair and efficient priority algorithm. Once the scheduler decides to serve a mobile, the mobile is served at the rate it requested over the DRC channel (the actual rate may be lower if the BS does not have enough information bits)

Upon receipt of the data rate request message, the base station adjusts the rate of a transmitted signal. The adjustments are performed for the next packet in response to information provided about the channel by a previous packet. A base station broadcasting at insufficient or excess data rates results

in reduced channel throughput or inefficient use of network resources, respectively.

Current implementations of the above technique however, have significant limitations. The SINR may change rapidly. The data rate that was appropriate for a previously transmitted packet may not be appropriate for a subsequently transmitted packet. The delay between the transmission of one packet and the generation and transmission of a data rate request message for a subsequent packet can result in reduced channel throughput, especially when the channel is characterized by rapid fluctuations in noise or other interference.

Hence a need exists in the art for an efficient system and method for maximizing communications system throughput that accounts for a changing SINR occurring between the determination of the rate control signal based on a previous packet and the application of the rate control signal to a subsequent packet. There is a further need for a system for adjusting the data rate of a transmitted signal in accordance with the changing SINR.

SUMMARY OF THE INVENTION

The need in the art is addressed by the system for providing an accurate prediction of a signal-to-interference noise ratio of the present invention. In the illustrative embodiment, the inventive system is employed in a wireless communications system and includes a first mechanism for receiving a signal transmitted across a channel via an external transmitter. A second mechanism generates a sequence of estimates of signal-to-interference noise ratio based on the received signal. A third mechanism determines a relationship between elements of the sequence of estimates. A fourth mechanism employs the relationship to provide a signal-to-interference noise ratio prediction for a subsequently received signal.

In the illustrative embodiment, the inventive system further includes a mechanism for generating a data rate request message based on the signal-to-noise ratio prediction. A transmitter transmits the data rate request message to the external transceiver. The external transceiver includes rate control

circuitry for receiving the data rate request message and adjusting a transmission rate of the signal in response thereto.

In the specific embodiment, the relationship between elements of the sequence of estimates is based on an average of the elements of the sequence of estimates. The third mechanism includes a bank of filters for computing the average. The impulse responses of the transfer functions associated with each filter in the bank of filters are tailored for different fading environments. The different fading environments include one environment associated with a rapidly moving system, a second environment associated with a slowly moving system, and a third system associated with a system moving at a medium velocity.

A selection mechanism is connected to each of the filter banks and selects an output from one of the filters in the filter bank. The selected output is associated with a filter having a transfer function most suitable to a current fading environment. In the present specific embodiment, the largest output is selected from the outputs of the filter bank based on the smallest error standard deviation. The resulting accurate prediction of the signal-to-interference noise ratio facilitates generating accurate rate requests.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a diagram of a wireless communications system transceiver constructed in accordance with the teachings of the present invention and employing a signal to interference and noise ratio (SINR) predictor.

Fig. 2 is a more detailed diagram of the SINR predictor of Fig. 1.

Fig. 3 is a more detailed diagram of the SINR predictor of Fig. 2.

DESCRIPTION OF THE INVENTION

While the present invention is described herein with reference to illustrative embodiments for particular applications, it should be understood that the invention is not limited thereto. Those having ordinary skill in the art and access to the teachings provided herein will recognize additional

modifications, applications, and embodiments within the scope thereof and additional fields in which the present invention would be of significant utility.

CDMA systems generally employ one of two methods to transmit a known pilot signal together with an unknown data signal. The methods include the pilot or reference symbol assisted method and the pilot channel assisted method. In the pilot symbol assisted method, a pilot signal comprising known symbols is spread by a pseudo-noise (PN) sequence and inserted into a data sequence spread by the same PN sequence in preparation for transmission to one or more mobile stations. In the pilot channel assisted method, the pilot signal and the data signal are spread with two different PN sequences, which are then added together and transmitted.

Fig. 1 is a diagram of a wireless communications system transceiver 10 of the present invention employing a signal to interference and noise ratio (SINR) predictor 12. The system 10 represents a CDMA mobile station. Signals received by the transceiver system 10 are received over a forward communications link between a base station (not shown) and the system 10. Signals transmitted by the transceiver system 10 are transmitted over a reverse communications link from the transceiver system 10 to the associated base station.

For clarity, many details of the transceiver system 10 have been omitted, such as clocking circuitry, microphones, speakers, and so on. Those skilled in the art can easily implement the additional circuitry without undue experimentation.

The transceiver system 10 is a dual conversion telecommunications transceiver and includes an antenna 14 connected to a duplexer 16. The duplexer 16 is connected to a receive path that includes, from left to right, a receive amplifier 18, a radio frequency (RF) to intermediate frequency (IF) mixer 20, a receive bandpass filter 22, a receive automatic gain control circuit (AGC) 24, and an IF-to-baseband circuit 26. The IF-to-baseband circuit 26 is connected to a baseband computer 28 at a despreading/decovering circuit 64 in a baseband computer 28.

The duplexer 16 is also connected to a transmit path 66 that includes a transmit amplifier 30, an IF-to-RF mixer 32, a transmit bandpass filter 34, a transmit AGC 36, and a baseband-to-IF circuit 38. The transmit baseband-to-IF circuit 38 is connected to the baseband computer 28 at an encoder 40.

Outputs of the despreading/decovering circuit 64 in the baseband computer 28 are connected to an SINR circuit 66 and a path weighting and combining circuit 42. Outputs of the SINR circuit 66 are connected to the SINR predictor 12, the LLR circuit 46, and the path weighting and combining circuit 42.

An input of a rate request generation circuit 44 is connected to an output of the SINR predictor 12. An output of the log-likelihood ratio (LLR) circuit 46 is connected to an input of a decoder 48, which is a turbo decoder in the present specific embodiment. An input of the LLR circuit 46 is connected to an output of the path weighting and combining circuit 42. An output of the decoder 48 is connected to an input of a controller 50 that is also connected to the rate request generation circuit 44 and to an input of the encoder 40.

The antenna 14 receives and transmits RF signals. A duplexer 16, connected to the antenna 14, facilitates the separation of receive RF signals 52 from transmit RF signals 54.

In operation, RF signals 52 received by the antenna 14 are directed to the receive path 64 where they are amplified by the receive amplifier 18, mixed to intermediate frequencies via the RF-to-IF mixer 20, filtered by the receive bandpass filter 22, gain-adjusted by the receive AGC 24, and then converted to digital baseband signals 56 via the IF-to-baseband circuit 26. The digital baseband signals 56 are then input to a digital baseband computer 28.

In the present embodiment, the receiver system 10 is adapted for use with quadrature phase shift-keying (QPSK) spreading and despreading techniques, and the digital baseband signals 56 are quadrature amplitude modulation (QAM) signals that include both in-phase (I) and quadrature (Q) signal components. The I and Q baseband signals 56 represent both pilot signals and data signals transmitted from a CDMA telecommunications transceiver such as a transceiver employed in a base station.

In the transmit path 66, digital baseband computer output signals 58 are converted to analog signals via the baseband-to-IF circuit 38, mixed to IF signals, filtered by the transmit bandpass filter 34, mixed up to RF by the IF-to-RF mixer 32, amplified by the transmit amplifier 30 and then transmitted via the duplexer 16 and the antenna 14.

Both the receive and transmit paths 64 and 66, respectively, are connected to the digital baseband computer 28. The digital baseband computer 28 processes the received baseband digital signals 56 and outputs the digital baseband computer output signals 58. The baseband computer 28 may include such functions as signal to data conversions and/or vise versa.

The baseband-to-IF circuit 38 includes various components (not shown) such as digital-to-analog converters (DACs), mixers, adders, filters, shifters, and local oscillators. The baseband computer output signals 58 include both in-phase (I) and quadrature (Q) signal components that are 90° out of phase. The output signals 58 are input to digital-to-analog converters (DACs) (not shown) in the analog baseband-to-IF circuit 38, where they are converted to analog signals that are then filtered by lowpass filters (not shown) in preparation for mixing. The phases of the output signals 58 are adjusted, mixed, and summed via a 90° shifter (not shown), baseband-to-IF mixers (not shown), and an adder (not shown), respectively, included in the baseband-to-IF circuit 38.

The adder outputs IF signals to the transmit AGC circuit 36 where the gain of the mixed IF signals is adjusted in preparation for filtering via the transmit bandpass filter 34, mixing up to RF via the IF-to-transmit mixer 32, amplifying via the transmit amplifier 20, and eventual radio transmission via the duplexer 16 and the antenna 14.

Similarly, the IF-to-baseband circuit 26 in the receive path 64 includes circuitry (not shown) such as analog-to-digital (ADC) converters, oscillators, and mixers. A received gain-adjusted signals output from the receive AGC circuit 24 are transferred to the IF-to-baseband circuit 26 where they are mixed to baseband via mixing circuitry and then converted to digital signals via analog-to-digital converters (ADCs) (not shown).

Both the baseband-to-IF circuit 38 and the IF-to-baseband circuit 36 employ an oscillator signal provided via a first oscillator 60 to facilitate mixing functions. The receive RF-to-IF mixer 20 and the transmit IF-to-RF mixer 32 employ an oscillator signal input from a second oscillator 62. The first and second oscillators 60 and 62, respectively, may be implemented as phase-locked loops that derive output signals from a master reference oscillator signal (not shown).

Those skilled in the art will appreciate that other types of receive and transmit paths 64 and 66 may be employed instead without departing from the scope of the present invention. The various components such as amplifiers 18 and 30, mixers 20 and 32, filters 22 and 34, AGC circuits 24 and 36, and frequency conversion circuits 26 and 38 are standard components and may easily be constructed by those having ordinary skill in the art and access to the present teachings.

In the baseband computer 28, the received I and Q signals 56 are input to the despreading/decovering circuit 64 where a pilot channel comprising pilot signals and a data channel comprising data signals are extracted from the received I and Q signals 56. The pilot channel and the data channel are provided to the SINR circuit 66 and the path weighting and combining circuit 42 from the despreading/devovering circuit 64.

The SINR circuit 66 outputs a SINR signal comprising a sequence of SINR values, i.e., samples, to the SINR predictor 12 and the LLR circuit 46. The SINR circuit 66 also outputs the reciprocal of the interference energy ($1/N_t$) to the path weighting and combining circuit 42.

The despread and decovered data channel signal, provided by the despreading/decovering circuit 64 to the path weighting and combining circuit 42 is also provided to the decoder 48 where it is decoded and forwarded to the controller 50. At the controller 50, the decoded signal is processed to output voice or data, or to generate a reverse link signal for transfer to the associated base station (not shown).

The path weighting and combining circuit 42 computes optimal ratio path combining weights for multipath components of the received data signal

corresponding to the data channel signal, weights the appropriate paths, combines the multiple paths, and provides the summed and weighted paths as a metric to the LLR circuit 46.

The LLR circuit 46 employs metrics from the path weighting and combining circuit 42 with the SINR estimation provided by the SINR circuit 66 to generate an optimal LLR and soft decoder decision values. The constructions of applicable LLR circuits are known in the art. In a preferred implementation, the LLR circuit 46 is constructed in accordance with the teachings of co-pending U.S. Patent Application Serial No. 09/311,793 filed May 13, 1999 entitled SYSTEM AND METHOD FOR PERFORMING ACCURATE DEMODULATION OF TURBO-ENCODED SIGNALS VIA PILOT ASSISTED COHERENT DEMODULATION, assigned to the assignee of the present invention and incorporated herein by reference.

The optimal LLR value is provided to the decoder 48 to facilitate decoding of the received data channel signals. The controller 50 then processes the decoded data channel signals to output voice or data via a speaker or other device (not shown). The controller 50 also controls the sending of speech signals and data signals from an input device (not shown) to the encoder 40 in preparation for transmission.

The rate request generation circuit 44 generates a rate control message based on the predicted SINR value for the next packet as provided by the SINR predictor 12. The SINR predictor 12 employs a filter bank (as discussed more fully below) to facilitate SINR prediction, which enables the rate request generation circuit 44 to provide accurate rate control messages.

The rate request generation circuit 44 compares the predicted SINR with a set of predetermined thresholds. The rate request generation circuit 44 generates a rate control request message based on the relative magnitude of the predicted SINR signal with respect to the various thresholds. The exact details of the rate request generation circuit 44 are application-specific and easily determined and implemented by those ordinarily skilled in the art to suit the needs of a given application.

The rate request generation circuit 44 subsequently provides a rate control message, also termed rate request message, which is transferred to the controller 50. The controller 50 prepares the rate request message for encoding via the encoder 40 and eventual transmission to the associated base station (not shown) over a data rate request channel (DRC) via the transmit path 66, duplexer 16 and antenna 14. When the base station receives the rate request message, the base station adjusts the rate of the transmitted signals accordingly.

The accurate SINR estimates and the total interference noise chip energy N_t estimates from the SINR circuit 66 improve the performance of the rate request generation circuit 44 and improve the performance of the decoder 48, thereby improving the throughput and efficiency of the transceiver system 10 and associated telecommunications system.

SINR estimation circuits are known in the art. In a preferred implementation, the SINR circuit 66 is constructed in accordance with the teachings of co-pending U.S. Patent Application Serial No. 09/310,053 filed May 11, 1999 SYSTEM AND METHOD FOR PROVIDING AN ACCURATE ESTIMATION OF RECEIVED SIGNAL INTERFERENCE FOR USE IN WIRELESS COMMUNICATIONS SYSTEMS, assigned to the assignee of the present invention and incorporated herein by reference.

The transceiver 10 of Fig. 1 is easily adapted for use in a base station instead of a mobile station, in which case, the transceiver 10 will contain rate and power adjustment functionality built into software running on the controller 50. The appropriate software is easily constructed by those having ordinary skill in the art and access to the present teachings.

While in the present specific embodiment, the predictor 12 provides SINR predictions to the rate request generation circuit 44, those skilled in the art will appreciate that the SINR predictions may be employed by another type of circuit such as a power control circuit without departing from the scope of the present invention.

Fig. 2 is a more detailed diagram of the SINR predictor 12 of Fig. 1. The SINR predictor 12 includes a sliding window averaging filter 70 that

receives SINR samples from the SINR circuit 66 of Fig. 1 as input. A SINR samples decibel converter and filter 72 also receives the SINR samples as input.

An output of the averaging filter 70 is connected to an input of a filter output decibel converter 74. An output of the decibel converter 74 is connected, in parallel, to an input of a fast fading SINR predictor 76, an input of a slow fading SINR predictor 78, and an input of a hold predictor 80. Outputs of the fast fading SINR predictor 76, the slow fading SINR predictor 78, and the hold predictor 80 are connected to a prediction selector 82. Another output of the fast fading SINR predictor 76 is connected, in parallel, to an input of the slow fading SINR predictor 78 and to an input of the hold predictor 80. An output of the SINR samples decibel converter and filter 72 is connected, in parallel, to an input of the slow fading SINR predictor 78 and an input of the hold predictor 80.

In operation, the averaging filter 70 and the SINR samples decibel converter and filter 72 receive the SINR samples from the SINR circuit 66 of Fig. 1. The averaging filter 70 computes an average of the received SINR samples over a predetermined number of samples. The predetermined number of samples is application-specific and easily determined by those ordinarily skilled in the art to meet the needs of a given application.

The averaged SINR samples output from the averaging filter 70 are converted to decibel scale via the filter output decibel converter 74. The resulting filtered decibel scale SINR samples are then provided, in parallel, to the fast fading SINR predictor 76, the slow fading SINR predictor 78, and the hold predictor 80.

The SINR samples decibel converter and filter 72 filters the received SINR samples and produces the decibel values of the SINR samples as output, the mean of the decibel values being adjusted to zero. The SINR sample decibel converter and filter 72 is application-specific and easily determined by those ordinarily skilled in the art. The resulting converted and filtered samples are provided to the slow fading SINR predictor 78 and the hold predictor 80.

The fast fading SINR predictor 76, the slow fading SINR predictor 78, and the hold predictor 80 form a filter bank. In fast fading signal environments, the fast fading SINR predictor 76 is designed to produce the smallest standard deviation of prediction error as output. Similarly, during slow fading signal environments, the slow fading SINR predictor 78 produces the smallest standard deviation of prediction error as output, and during medium fading signal environments, the hold predictor 80 produces the smallest standard deviation of prediction error as output.

The prediction selector 82 selects from the outputs of the SINR predictors 76, 78, and 80 the signal having the smallest standard deviation of prediction error value, which is most representative of the current fading signal environment. The selected prediction is output from the prediction selector 82, which is easily implemented by those ordinarily skilled in the art. The outputs of the SINR predictors 76, 78, and 80, are backed-off by predetermined factors to prevent overshooting of the SINR prediction as discussed more fully below.

Those skilled in the art will appreciate that a single filter having transfer function coefficients that are selectively changed in accordance with changing fading signal environments may be used in place of the filter bank comprising the SINR filters 76, 78, and 80 without departing from the scope of the present invention. In addition, different filter coefficients and/or additional filters may be employed, without departing from the scope of the present invention.

The SINR predictors 76, 78, and 80 are linear prediction filters and are designed to mimic Wiener filter behavior.

In general, a signal y(n) often contains a signal component x(n) and a noise component w(n) such that y(n) = x(n) + w(n) where n is the sample number. A desired signal is always a linear function of x(n) and can be estimated from y(n). In the present case, x(n) represents the SINR samples.

Prediction is a special case of estimating the desired signal is in advance of a current observation. The desired signal d(n + D) is D samples ahead of y(n), where D is a predetermined number and is greater than or

equal to 5 samples in the present embodiment. The difference between a prediction $\hat{d}$ (n) of the desired signal d(n) and the desired signal d(n) is an error e(n). It is well known in the art that the optimum linear filter is a Wiener filter in the sense that it results in a minimum mean-square error.

The desired signal d(n) in the present embodiment is the average SINR over the packet length. Different packet lengths correspond to different desired signals. The transceiver 10 of Fig. 1 runs the predictions for five different packet sizes (1, 2, 4, 8, and 16 slot packets). Upon receipt of a path combined SINR estimate, which is updated every half-slot, the transceiver 10 of Fig. 1 (which corresponds to a mobile station) runs the predictor 12 five times corresponding to the packet sizes of {1,2,4,8,16) slots, respectively. Hence, the predictor 12 updates the processing shown in Fig 3 five times for five different packet lengths with different values of parameters like prediction delay and filter coefficients.

Fig. 3 is a more detailed diagram for the SINR prediction of a given packet length implemented via the SINR predictor 12 of Fig. 2. The SINR samples decibel converter and filter 72 includes a first decibel converter 90, an input of which receives the SINR samples from the SINR circuit 66 of Fig. 1, and an output of which is connected to a positive terminal of a subtractor 92 and an input of a filter (F_1) 96. An output of the filter 96 is connected to a negative terminal of the first subtractor 92.

In operation, the SINR samples decibel converter and filter 72 converts the received SINR samples to decibel scale via the decibel converter 90 and filters the decibel signals via the first filter 96. The filtered decibel samples are subtracted from the decibel samples output from the decibel converter 90. The output of the SINR samples decibel converter and filter 72 is described by the following equation:

$$u_o(n) = u(n) - m_u(n), \qquad [1]$$

where $u_o(n)$ represents the output samples of the SINR samples decibel converter and filter 72; u(n) represents the decibel-scale samples output from

the decibel converter 90; and $m_u(n)$ represents the mean of the decibel-scale samples output from the first filter 96.

The transfer function $F_1(z)$ of the first filter 96 is described by the following equation:

$$F_1(z)=\frac{\lambda}{1-(1-\lambda)z^{-1}}, \quad [2]$$

where λ is a constant coefficient and z is a complex variable. The coefficient λ is application-specific and is easily determined by those ordinarily skilled in the art to meet the needs of a given application.

The received SINR samples from the SINR circuit 66 of Fig. 1 are also input to the sliding window averaging filter 70. The averaging filter 70 computes an average of the SINR samples over L samples, where L represents the given packet length.

An output of the averaging filter 70 is connected to the filter output decibel converter 74 that converts the output of the averaging filter 70 to decibel scale in accordance with methods well known in the art. The resulting decibel values which represent the desired signal are input to the fast fading SINR predictor 76, the slow fading SINR predictor 78, and the hold predictor 80.

In the fast fading SINR predictor 76, the output of the filter output decibel converter 74 is connected to a negative terminal of a second subtractor 106. An output of the decibel converter 74 is connected to a filter (F_3) 100. An output of the filter 100 is connected to a first delay 102, a first back-off circuit 104, and to a first adder 120 and a second adder 150 in the hold predictor 80 and the slow fading SINR predictor 78, respectively. An output of the first back-off circuit 104 is connected to an input of the prediction selector 82. A second input of the second subtractor circuit 106 is connected to an output of the first delay 102. An output of the second subtractor circuit 106 is connected to a first squaring circuit 108, which has an output connected to an input of the first filter (F_4) 112. An output of the filter 112 is connected to an input of a

first square root circuit 114. An output of the first square root circuit 114 is connected to an input of the first back-off circuit 104.

In operation, the fast fading SINR predictor 76 receives the decibel-scale samples from the filter output decibel converter 74 at the filter F_3 100 and at a negative terminal of the second subtractor 106. The filter F_3 100 computes a long-term average of the decibel values and is represented by the following equation:

$$m_d(n) = \hat{d}_1(n+D) = (1-\alpha)m_d(n-1) + \alpha d(n), \quad [3]$$

where $m_d(n)$ is the long term mean of the received decibel-scale samples at a particular sample n and represents a mean SINR prediction $\hat{d}_1(n+D)$, which is D samples in the future, where D is a predetermined delay based on the given packet length. α is a predetermined coefficient of the transfer function (F_3) of the filter 100; d(n) is the current output from decibel converter 74, and $m_d(n-1)$ long-term average one sample ago. The transfer function F3 of the filter 100 is also described by the following equation:

$$F_3(z) = \frac{\alpha}{1-(1-\alpha)z^{-1}}, \quad [4]$$

where z is a complex variable, and α is a predetermined coefficient as noted above. α is easily determined by those ordinarily skilled in the art to meet the needs of a given application.

An the resulting long-term average $m_d(n)$ output from the filter 100 is delayed by D samples via the first delay circuit 102 and provided to a positive terminal of the second subtractor 106. The second subtractor subtracts d(n) output from the filter output decibel converter 74 from the long-term average $m_d(n)$ and provides a prediction error signal $e_1(n)$ in response thereto. The resulting error signal $e_1(n)$ is squared and filtered by the squaring circuit 108 and the first filter F_4 112, respectively. The first filter F_4 112 is an infinite

impulse response (IIR) filter having a transfer function $F_4(z)$ described by the following equation:

$$F_4(z)=\frac{\beta}{1-(1-\beta)z^{-1}}, \quad [5]$$

where β is a filter coefficient, and the other variables are as described above.

The filtered, i.e., averaged, squared values are input to the square root circuit 114, which computes the square root mean square ($rmse_1$) of the error signal $e_1(n)$. The root means square error $rmse_1$ is provided to the first back-off circuit 104, where $rmse_1$ is multiplied by a predetermined constant k_1. The exact value for k_1 is application-specific and may be a constant or may be dynamically updated in accordance with a changing signal environment by another circuit (not shown) or software routine.

The root means square error $rmse_1(n)$ is described by the following equation:

$$rmse_1(n)=\sqrt{(1-\beta)\cdot mse_1(n-1)+\beta[e_1(n)]^2}, \quad [6]$$

where β is as given for equation (5); mean square error $mse_1(n-1)$ represents the output from the first filter F_4 112 one sample ago.

The first back-off circuit 104 reduces the first prediction $\hat{d}_1(n+D)$ by k_1*$rmse_1$ to reduce prediction over-shoot. The reduced first prediction is denoted $\hat{d}_1^{'}(n+D)$ and is described by the following equation:

$$\hat{d}_1^{'}(n+D)=\hat{d}_1(n+D)-k_1\cdot rmse_1(n), \quad [7]$$

where the variables are as given above.

The structures of the slow-fading SINR predictor 78 and the hold predictor 80 are similar to the structure of the fast fading SINR predictor 76. However, the slow-fading SINR predictor 78 includes an additional filter F_2 116, and the first adder 150. The hold predictor 80 includes an additional hold

filter 118 and a second adder 120. The first adder 150 and the second adder 120 receive the long-term mean $m_d(n)$ output from the filter F_3 100 of the fast fading SINR predictor 76.

The slow fading SINR predictor 78, from left to right and from top to bottom, a third subtractor 122, a second squaring circuit 124, a second filter F_4 128, a second square root circuit 130, the filter F_2 116, the first adder 150, a second delay 132, and a second back-off circuit 134.

In operation, the filter F_2 116 filters output from the SINR samples decibel converter and filter 72. The transfer function $F_2(z)$ of the second filter F_2 116 is described by the following equation:

$$F_2(z)=\frac{\mu}{1-(1-\mu)z^{-1}}, \qquad [8]$$

where μ a predetermined filter coefficient. The output $\hat{d}_0(n+D)$ of the second filter F_2 116 is described by the following equation:

$$\hat{d}_0(n+D)=(1-\mu)\hat{d}_0(n+D-1)+\mu u_o(n), \qquad [9]$$

where μ is as given above; $\hat{d}_0(n+D-1)$ is the output $\hat{d}_0(n+D)$ delayed by one sample; and $u_0(n)$ is the output of the SINR samples decibel converter and filter 72.

The output of the filter F_2 116 as described by equation (9) is input to a terminal of the first adder 150, which adds the output of the long-term average $m_d(n)$, which is provided from the fast fading SINR predictor 76. The resulting sum is denoted $\hat{d}_2(n+D)$ and is described by the following equation:

$$\hat{d}_2(n+D)=\hat{d}_0(n+D)+m_d(n), \qquad [10]$$

where the variables are as given above.

The output of the first adder 150 as given by equation (10) is input, in parallel, to the second delay 132 and the second back-off circuit 134. The delay 132 delays the output of the first adder 150 by D and provides the result to a positive terminal of the third subtractor 122. The third subtractor subtracts the output d(n) of the filter output decibel converter 74 from the delayed result to yield a second error signal $e_2(n)$, which is described by the following equation:

$$e_2(n) = \hat{d}_2(n) - d(n), \qquad [11]$$

where $\hat{d}_2(n)$ is the delayed output of the first adder 150, i.e., is the output of the second delay 132, and d(n) is the output of the filter output decibel converter 74.

The resulting error signal $e_2(n)$ is squared and filtered by the second squaring circuit 124 and the second filter F_4 128, respectively. The transfer function of the second filter F_4 128 is as described in equation (5). The square root of the output of the filter F_4 128 is computed by the second square root circuit 130 and yields the following output:

$$rmse_2(n) = \sqrt{(1-\beta)mse_2(n-1) + \beta[e_2(n)]^2}, \qquad [12]$$

where $rmse_2(n)$ is the root mean square error of the signal $e_2(n)$; mean square error $mse_2(n-1)$ is the output of second filter F_4 128 delayed by one sample; and the other variables and constants are as given above.

The resulting root means square error $rmse_2(n)$ is multiplied by a predetermined factor k, and the result is subtracted from the output of the first adder 150 to yield the following output:

$$\hat{d}_2'(n+D) = \hat{d}_2(n+D) - k_2 \cdot rmse_2(n), \qquad [13]$$

where the constants and variables are as described above. The output $d_2'(n+D)$ of the second back-off circuit 134 is provided to the prediction selector 82.

The predetermined factor k_2 that is application specific and easily determined by those ordinarily skilled in the art. The factor k_2 may be equivalent to the factors k_1 and k_3 employed in the first back-off circuit 104 and the third back-off circuit 148 and may be dynamically altered without departing from the scope of the present invention.

The hold predictor 80 includes, from left to right and from top to bottom, a fourth subtractor 136, a third squaring circuit 138, a third filter F_4 142, a third square root circuit 144, a third delay circuit 146, the hold filter circuit 118, the second adder 120, and a third back-off circuit 148.

In the present specific embodiment, the hold predictor 80 is only employed when the packet length is less than or equal to 2 slots. The hold predictor 80 is selectively activated by a circuit (not shown) that determines when the packet length is less than or equal to 2 slots and selectively enables the output of the hold predictor 80.

In operation, the hold filter circuit 118 filters the output of the SINR samples decibel converter and filter 72 and provides the result to a terminal of the second adder 120, which adds the output $m_d(n)$ of the filter 100 of the fast fading SINR predictor 76. The output of the adder 120 is described by the following equation:

$$\hat{d}_3(n+D)=u_0(n)\cdot HoldWeight+m_d(n), \qquad [14]$$

where the HoldWeight is provided by the hold filter circuit 118, and $u_0(n)$ is the output of the SINR samples decibel converter and filter 72.

The resulting output is delayed by D samples by the third delay 146 to yield $\hat{d}_3(n)$. The output d(n) of the filter output decibel converter 74 is then subtracted from the delayed samples $\hat{d}_3(n)$ to yield a third error signal $e_3(n)$ described by the following equation:

$$e_3(n) = \hat{d}_3(n) - d(n), \qquad [15]$$

where the variables are as given above.

The subsequent third squaring circuit 138, the third filter F_4 142, and the third square root circuit 144 compute the root mean square error signal $rmse_3(n)$ of the error signal $e_3(n)$, which is described by the following equation:

$$rmse_3(n) = \sqrt{(1-\beta)mse_3(n-1) + \beta[e_3(n)]^2}\,, \qquad [16]$$

where mean square error $mse_3(n-1)$ is the output of third filter F_4 142 delayed by one sample, and the other constants and variables are as given above. The transfer function of the third filter F_4 142 is as given in equation (5).

The resulting root mean square error $rmse_3(n)$ is multiplied by the predetermined constant k_3 via the third back-off circuit 148. The result is subtracted from the output $\hat{d}_3(n+D)$ of the second adder 120 to yield the following output:

$$\hat{d}_3'(n+D) = \hat{d}_3(n+D) - k_3 \cdot rmse_3(n)\,, \qquad [17]$$

where the constants and variables are as given above. The result given by equation (17) is provided to the prediction selector circuit 82.

The prediction selector 82 chooses the prediction with the smallest rmse value as a final prediction for the given packet length. For 1 and 2 slot packets, the prediction selector 82 chooses from the fast fading predictor 76, the slow fading predictor 78, and the hold predictor 80. For 4, 8, and 16 slot packets, the prediction selector 82 chooses from the fast filter 76 and the slow fading filter 78.

The delays 102, 132, and 146 provide delays of D half slots, where D is prediction latency for the given packet length. The predictor 12 receives SINR

estimation samples once every half slot but only produces packet average SINR predictions once every two half slots. In addition, the filter F_1 96 is applied once every half slot, the filters 100, 112, 116, 128, and 142 having the transfer functions F_2, F_3, and F_4 are applied once every 2 half slots. The descriptions of the transfer functions $F_1(z)$, $F_2(z)$, $F_3(z)$, and $F_4(z)$ neglect the effects of decimation processing. However, those having ordinary skill in the art can easily adjust the transfer functions accordingly.

Those skilled in the art will appreciate that the SINR predictor 12 may be implemented in software without departing from the scope of the present invention, in which case, the filters 96, 100, 112, 128, 142, and 116 are easily switched on or off in accordance with the above rules.

Thus, the present invention has been described herein with reference to a particular embodiment for a particular application. Those having ordinary skill in the art and access to the present teachings will recognize additional modifications, applications, and embodiments within the scope thereof.

It is therefore intended by the appended claims to cover any and all such applications, modifications and embodiments within the scope of the present invention.

What is claimed is:

CLAIMS

1. A system for providing an accurate prediction of a signal-to-interference noise ratio comprising:

means for receiving a signal transmitted across a channel via a transmitter;

means for generating a sequence of estimates of signal-to-interference noise ratio based on said received signal;

means for determining a relationship between estimates of said sequence of estimates; and

means for employing said relationship to provide a signal-to-interference noise ratio prediction for a subsequently received signal.

2. The system of Claim 1 further including means for generating a data rate request message based on said signal-to-noise ratio prediction.

3. The system of Claim 2 further including means for transmitting said data rate request message to said transmitter.

4. The system of Claim 1 wherein said relationship is based on an average of elements of said sequence of estimates.

5. The system of Claim 4 wherein said means for determining includes a bank of filters for computing said average.

6. The system of Claim 5 wherein said bank of filters includes finite impulse response filters.

7. The system of Claim 5 wherein coefficients of transfer functions associated with each filter in said bank of filters are tailored for different fading environments.

8. The system of Claim 7 wherein said different fading environments include different Raleigh fading environments, one environment associated with a rapidly moving system, a second environment associated with a slowly moving system, and a third system associated with a system moving at a medium velocity.

9. The system of Claim 7 further including a selection circuit connected to each of said filter banks for selecting an output from one of said filters in said filter bank, said output associated with a filter having a transfer function most suitable to a current fading environment.

10. An efficient communications system transceiver for providing accurate control signals to an external transceiver in accordance with interference characteristics of a channel over which said communications system transceiver and said external transceiver communicate comprising:

an antenna;

a duplexer in communication with said antenna;

a receive path in communication with said duplexer for receiving a signal over said channel and providing a digital signal in response thereto;

a baseband computer for receiving and processing said digital signal;

a signal to interference and noise ratio predictor for providing a signal to interference and noise ratio prediction for a subsequently received digital signal; and

rate request means for generating a rate control message and providing said rate control message to said external transceiver in response to said signal to interference and noise ratio prediction.

11. The transceiver of Claim 10 wherein said rate request means is implemented in said baseband computer.

12. The transceiver of Claim 11 wherein said rate request means includes means for despreading said digital signal, estimating a carrier signal to interference value for said despread digital signal.

13. The transceiver of Claim 12 wherein said signal to interference and noise ratio predictor includes a predictor filter bank, each filter in said predictor filter bank having a signal to interference and noise ratio predictor for predicting a signal to interference and noise ratio in accordance with a predetermined fading characteristic of said channel.

14. The transceiver of Claim 13 wherein said signal to interference and noise ratio predictor further includes a prediction selector for selecting from each of said signal to interference and noise ratio predictors a signal to interference and noise ratio most suitable to a current fading characteristic of said channel.

15. A system for accurately predicting a signal to interference and noise ratio for a signal received over a channel comprising:

first means for providing carrier signal to noise ratio values from said received signal;

second means for filtering said carrier signal to noise ratio values in accordance with fading characteristics of said channel and providing outputs in response thereto for a slow fading, medium fading, and a fast fading channel; and

third means for selecting from said outputs a predicted signal to interference and noise ratio.

16. The system of Claim 15 wherein said second means includes back-off circuitry for adjusting each of said outputs by a predetermined factor.

17. The system of Claim 15 wherein said second means includes an averaging filter for providing an average of said carrier signal to noise ratio values.

18. The system of Claim 17 wherein said averaging filter is connected to a decibel conversion circuit for converting said average of said carrier signal to noise ratio values to decibel scale and providing decibel scale values in response thereto.

19. The system of Claim 18 wherein said second means includes a filter bank that includes a first filter, said first filter for receiving said decibel scale values and providing a signal to interference and noise ratio prediction based on a slow fading channel.

20. The system of Claim 19 wherein said first filter includes a decimation circuit for decimating said decibel scale values and providing decimated values in response thereto.

21. The system of Claim 20 wherein said first filter includes a long-term averaging filter for averaging said decimated values and providing said averaged values to a delay circuit, an output of said delay circuit connected to a subtractor circuit for subtracting delayed decimated values from said decibel scale values and providing a subtractor output in response thereto to a squaring circuit in communication with an additional averaging filter and a square root circuit, said square root circuit providing output to a first back-off circuit, first back-off circuit also receiving as input said averaged values and providing said signal to interference and noise ratio based on said slow fading channel in response thereto.

22. The system of Claim 21 wherein said filter bank further includes a second filter and a third filter for providing a carrier signal to interference

predictions suitable for a slow fading channel and a medium fading channel, respectively.

23. The system of Claim 15 wherein said second means for filtering includes a filter bank having a filter for each of said slow fading, medium fading, and fast fading channel characteristics.

24. The system of Claim 23 wherein said filter bank includes a first filter for approximating a Wiener filter, said first filter providing a signal to interference and noise ratio prediction for a slow fading channel.

25. The system of Claim 24 wherein said filter bank includes a second filter for approximating a Wiener filter, said second filter providing a signal to interference and noise ratio prediction for a fast fading channel.

26. The system of Claim 25 wherein said filter bank includes a third filter for providing a signal to interference and noise ratio prediction for a medium fading channel, said third filter implemented as a hold filter.

27. A system for accurately predicting a signal to interference and noise ratio for a signal received over a channel comprising:

first means for obtaining signal to noise ratio values based on said received signal;

second means for filtering said carrier signal to noise ratio values in accordance with a fast fading channel and providing a fast fade carrier signal to noise ratio prediction in response thereto;

third means for filtering said carrier signal to noise ratio values in accordance with a slow fading channel and providing a slow fade carrier signal to noise ratio prediction in response thereto;

fourth means for filter in said carrier signal to noise ratio values in accordance with a medium fading channel and providing a medium fade carrier signal to noise ratio prediction in response thereto; and

fifth means for selecting from said fast fade, slow fade, and medium fade carrier signal to noise ratio predictions and providing a predicted signal to interference and noise ratio in response thereto.

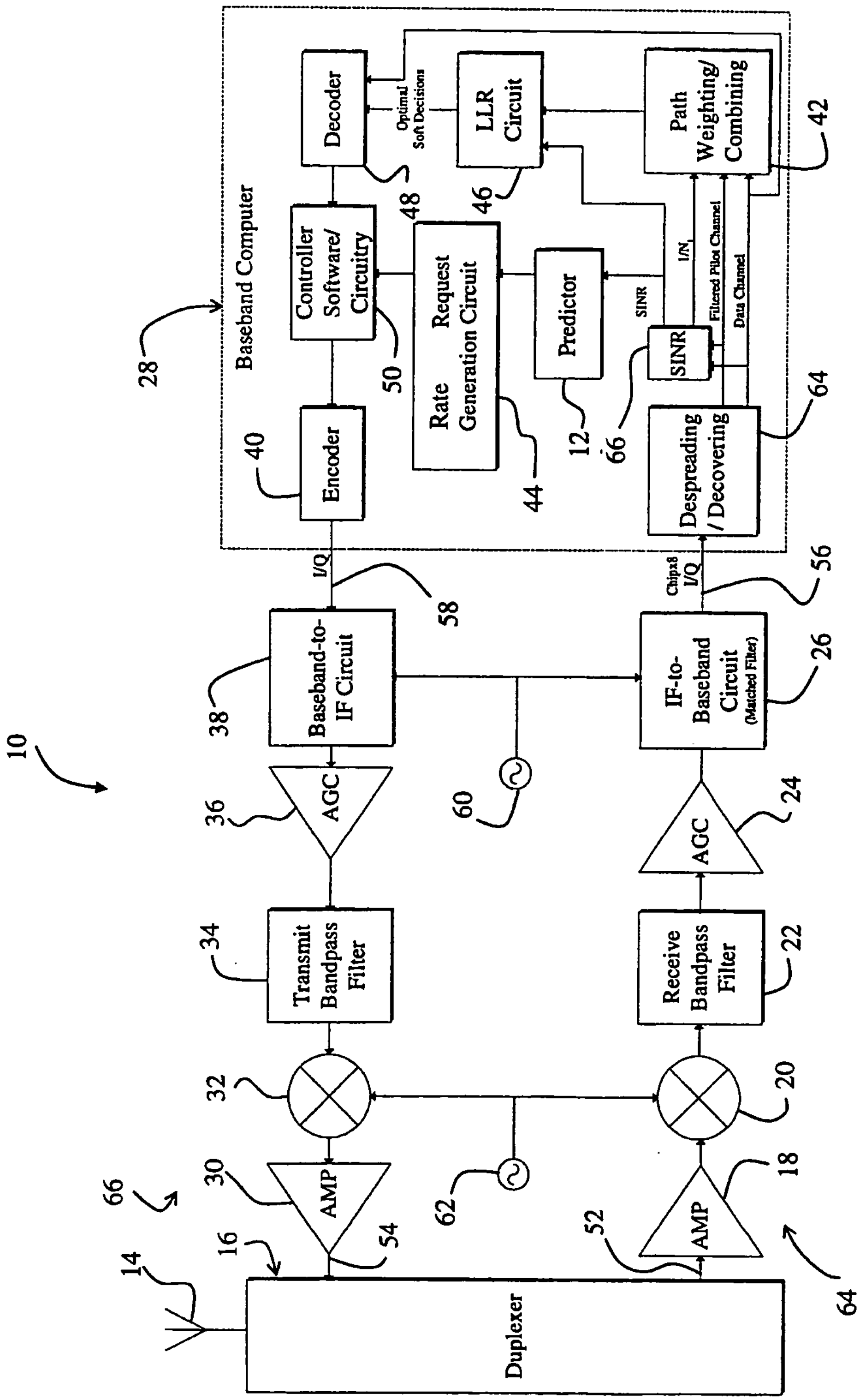

FIG. 1 Mobile Station

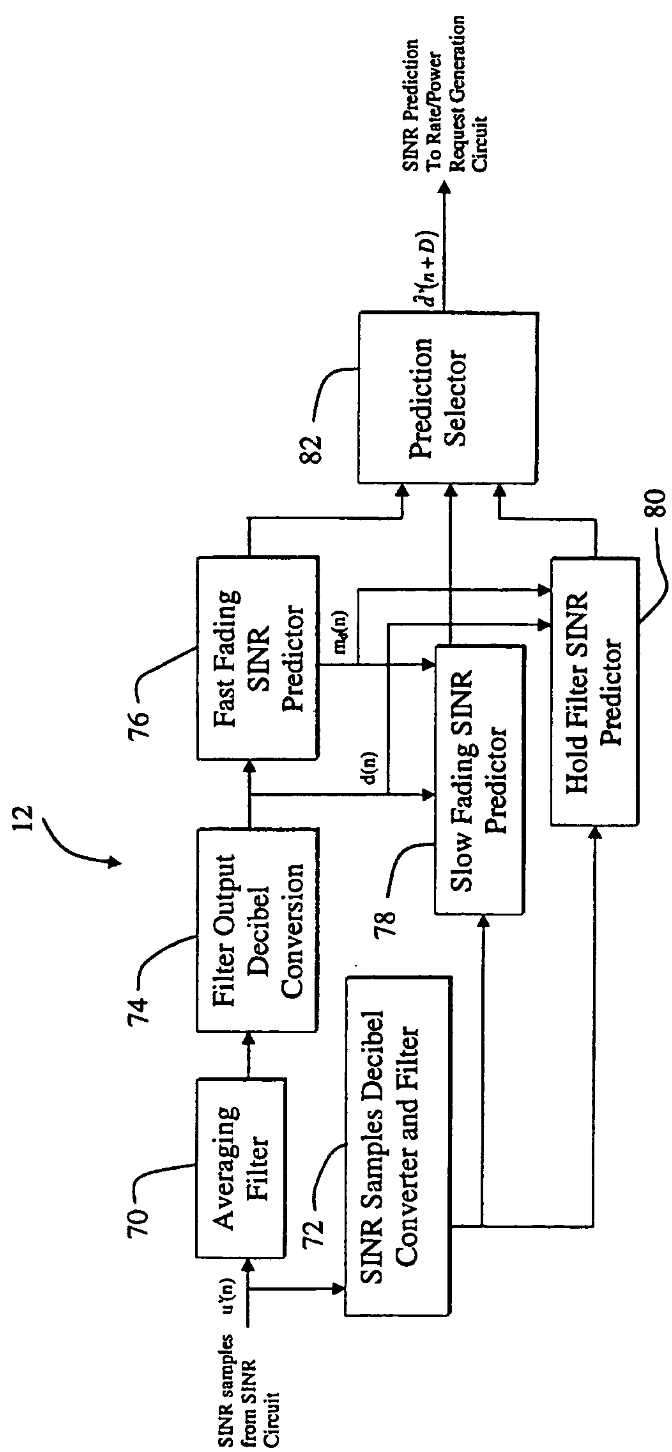
12
SINR samples from SINR Circuit
u'(n)
70
Averaging Filter
72
SINR Samples Decibel Converter and Filter
74
Filter Output Decibel Conversion
76
Fast Fading SINR Predictor
d(n)
$m_d(n)$
78
Slow Fading SINR Predictor
80
Hold Filter SINR Predictor
82
Prediction Selector
$\hat{d}'(n+D)$
SINR Prediction To Rate/Power Request Generation Circuit

FIG. 2

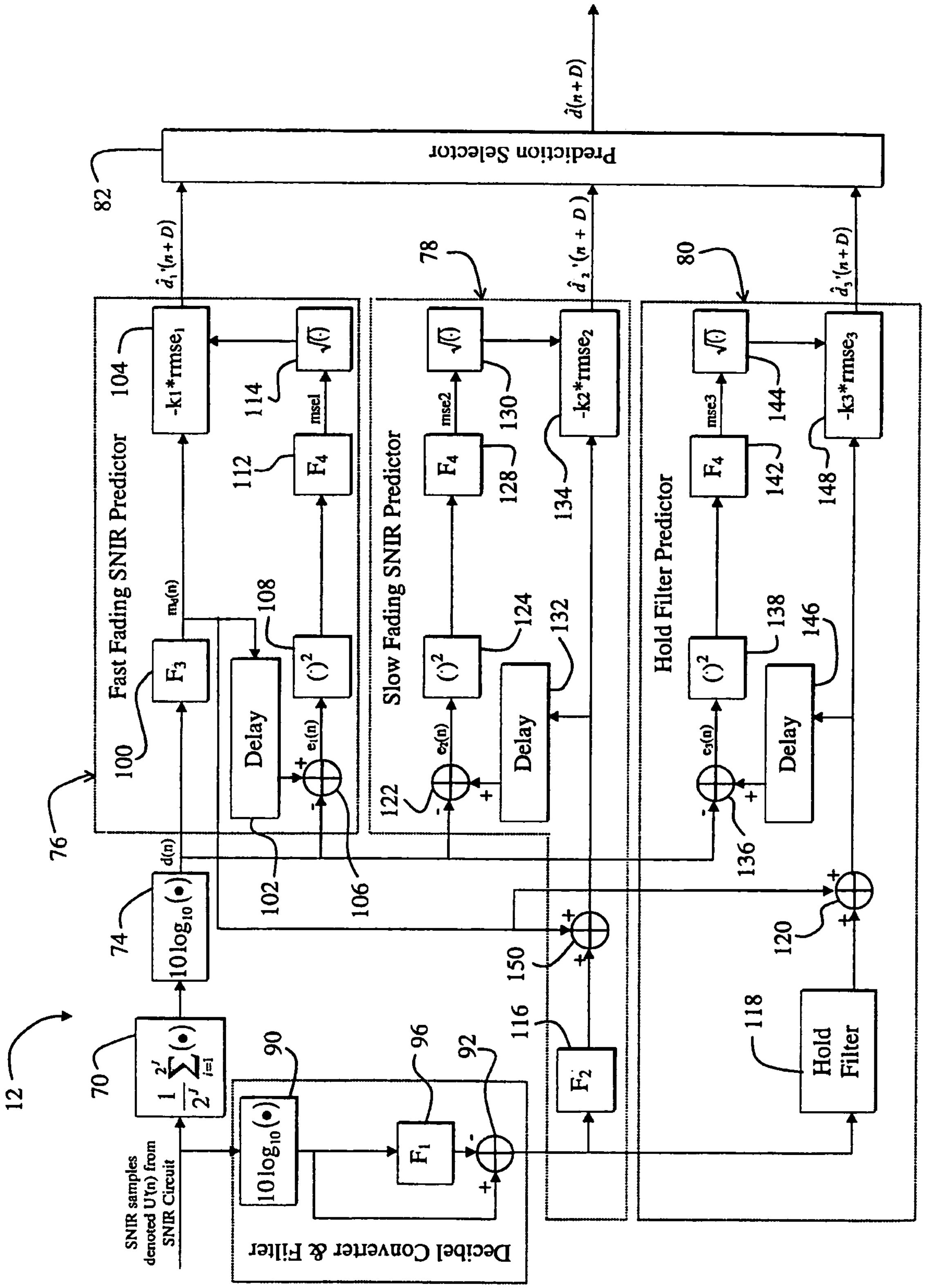
12
SNIR samples denoted U'(n) from SNIR Circuit
70
74
10log10(•)
d(n)
76
Fast Fading SNIR Predictor
100
F3
md(n)
Delay
102
e1(n)
106
108
( )2
112
F4
mse1
114
104
-k1*rmse1
78
Slow Fading SNIR Predictor
122
e2(n)
Delay
132
124
( )2
128
F4
mse2
130
134
-k2*rmse2
80
Hold Filter Predictor
136
e3(n)
Delay
146
138
( )2
142
F4
mse3
144
148
-k3*rmse3
Decibel Converter & Filter
90
10log10(•)
F1
96
92
116
F2
150
118
Hold Filter
120
82
Prediction Selector

FIG. 3

INTERNATIONAL SEARCH REPORT

Inter   nal Application No
PCT/US 00/24955

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04B1/10 H04B17/00 H04L1/20

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04B H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EPO-Internal, WPI Data, PAJ

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X A	EP 0 899 906 A (LUCENT TECHNOLOGIES INC) 3 March 1999 (1999-03-03) abstract page 1, line 45 - line 52 page 2, line 47 -page 3, line 5 page 8, line 32 -page 9, line 56 figure 2 figure 11 figure 12 ---	1-6, 10-12 7,13,15, 17,27
X A	US 5 541 955 A (JACOBSMEYER JAY M) 30 July 1996 (1996-07-30) abstract column 1, line 15 -column 12, line 24 figure 1 --- -/--	1-4, 10-12 5,13,15, 27

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

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"A" document defining the general state of the art which is not considered to be of particular relevance

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"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
21 December 2000	11/01/2001

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Lindhardt, U

1

INTERNATIONAL SEARCH REPORT

Inter nal Application No
PCT/US 00/24955

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
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INTERNATIONAL SEARCH REPORT

..ormation on patent family members

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